Configuring Survivable Remote Site Telephony

Survivable Remote Site Telephony (SRST) provides Cisco CallManager with fallback support for Cisco IP Phones that are attached to the Catalyst 4224 Access Gateway Switch (Catalyst 4224) on your local Ethernet.

SRST enables the Catalyst 4224 to provide call handling support for Cisco IP Phones when a Cisco IP Phones lose connection to the remote primary, secondary, or tertiary Cisco CallManager or when the WAN connection is down.

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Overview of Survivable Remote Site Telephony

Cisco CallManager Releases 3.0 and later support Cisco IP Phones across the WAN that are attached to a branch office Catalyst 4224. Without the SRST feature, when the WAN connection between the branch office Catalyst 4224 and Cisco CallManager fails or when connectivity with Cisco CallManager is lost, the Cisco IP Phones at the remote site are unusable for the duration of the failure.

The SRST feature overcomes this problem, enabling the basic features of the Cisco IP Phones by providing call-handling support on the Catalyst 4224 for its attached Cisco IP Phones. The system automatically detects the failure and uses the Simple Network Auto Provisioning (SNAP) technology to autoconfigure the Catalyst 4224 to provide call processing for the local Cisco IP Phones.

When the WAN link or connection to the primary Cisco CallManager is restored, call-handling capabilities for the Cisco IP Phones switch back to the primary Cisco CallManager. During a failure when the SRST feature is enabled, the Cisco IP Phone displays a message stating that the Cisco IP Phones are in the Cisco CallManager fallback mode and are able to perform limited functions.

Note The Cisco IP Phone 7960 and 7940 displays the message “CM Fallback Service Operating.” The Cisco IP Phone 7910 displays the message “CM Fallback Service.”

Restrictions

Restrictions on the SRST feature are as follows:

- Supports only the Cisco IP Phone 7960, Cisco IP Phone 7940, and Cisco IP Phone 7910 models

Note First-generation Cisco IP Phones, such as Cisco IP Phone 30 VIP and Cisco IP Phone 12 SP+ are not supported.

- Does not support other Cisco CallManager applications or services such as Cisco IP SoftPhone, Cisco Unity, or Cisco IP Contact Center
- Supports 24 Cisco IP Phones on the Catalyst 4224

Prerequisites

Prerequisites for employing the SRST feature on the Catalyst 4224 are as follows:

- IP routing must be enabled.
- The SRST Catalyst 4224 must be configured as the default for the Cisco IP Phones.
- Cisco IOS Release 12.1(5)YD or higher is required.
- Cisco CallManager Release 3.0.5 or higher is required.
- You must use the appropriate Cisco IP Phone load versions that support the SRST feature for the Cisco IP Phone 7960, Cisco IP Phone 7940, and Cisco IP Phone 7910 models.

For further details, refer to your Cisco CallManager documentation.

Supported Features

The following features are supported on the Cisco IP Phones:

- Re-homing of Cisco IP Phones to use call processing on the local Catalyst 4224
- Cisco IP Phone and plain old telephone service (POTS) telephones on the Catalyst 4224
- Graying out of all Cisco IP Phone function keys that are not supported during SRST mode:
  - CFwdAll (call forward all)
  - MeetMe
  - PickUp
  - GPickUp (group pickup)
  - Park
  - Confrn (conference)
- Extension-to-extension dialing
- Direct inward dialing (DID)
- Direct outward dialing (DOD)

- Calling party ID (Caller ID/ANI) display
- Calling party name display
- Last number redial
- Maintaining local extension-to-extension calls when WAN link fails
- Maintaining local extension to public switched telephone network (PSTN) calls when WAN link fails
- Maintaining existing calls when failed WAN link is reestablished
- Call transfer of local calls (blind transfer)
- Multiple lines per Cisco IP Phone
- Multiple line appearance across telephones
- Call hold (shared lines)
- Analog Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) ports
- BRI support for EuroISDN
- PRI support for NET5 switch type
- Support for a maximum of 48 Cisco IP Phones

Note When the primary Cisco CallManager is restored, the displayed message is removed and you can access all the previous functionality of the Cisco IP Phone.

Fallback Behavior

When the Cisco IP Phones lose contact with all primary, secondary, and tertiary Cisco CallManagers, the phones re-home to the local SRST Catalyst 4224 to provide the call processing capability required to place and receive calls. (Re-homing is a major network change involving moving the local loop termination from one Central Office wire center to another.) The Cisco IP Phone lists the IP address of the local SRST Catalyst 4224 as the default Catalyst 4224 in the Network Configuration area of the Settings menu. This list supports up to five default Catalyst 4224 entries; however, Cisco CallManager uses a maximum

of three entries. When a secondary Cisco CallManager is not configured, the SRST Catalyst 4224 is listed as the stand-by Cisco CallManager during normal operation hosted by a single Cisco CallManager.

When the WAN link fails, calls in progress are sustained if possible for the duration of the call. Calls in transition must be attempted again after the Cisco IP Phones re-home to the local SRST Catalyst 4224. The telephone service is unavailable from the time the connection is lost from the remote Cisco CallManager until the Cisco IP Phone is re-homed to the Catalyst 4224 with the SRST feature.

The time taken to re-home to the remote Cisco CallManager depends in part on the keep-alive period set for Cisco CallManager. Typically, it takes three times the keep-alive period for the phone to discover that its connection to Cisco CallManager has failed. The default keep-alive period is 30 seconds.

If the phone has an active standby connection established with the SRST Catalyst 4224, the fallback process itself takes 10 to 20 seconds after the primary Cisco CallManager has failed. An active standby connection to the SRST Catalyst 4224 exists only if the phone has a single Cisco CallManager in its Cisco CallManager list. Otherwise, the phone will maintain a standby connection to its secondary Cisco CallManager.

If the phone has multiple Cisco CallManagers in its Cisco CallManagers list, it must search through the list of secondary and tertiary Cisco CallManagers, which means the fallback time increases. The phone attempts to connect to each of its alternate Cisco CallManagers in turn before attempting to connect to the SRST Catalyst 4224 as a last resort.

Each Cisco CallManager connection attempt will take about one minute. A message is displayed on the Cisco IP Phone indicating that the Cisco IP Phones are in Cisco CallManager fallback mode.

These telephone services are restricted to those Cisco IP Phones that are supported by SRST Catalyst 4224.

Note The Cisco IP Phone 7960 and 7940 display the message “CM Fallback Service Operating” at the bottom of the screen. The Cisco 7910 IP Phone displays the message “CM Fallback Service” for 5 seconds every 30 seconds. The message is brief because the Cisco IP Phone 7910 has only a two-line display area and the display panel is also used to display the telephone number.

Cisco IP Phones periodically attempt to reestablish connections with the remote Cisco CallManagers. When a connection is reestablished with a remote Cisco CallManager, the Cisco IP Phones unregister from the local Catalyst 4224 with the SRST feature and register with the remote Cisco CallManager. The connection to the primary Cisco CallManager at the remote office cannot be reestablished if the telephone has active calls.

Figure 8-1 shows a branch office with several Cisco IP Phones connected to a Catalyst 4224. The Catalyst 4224 is connected to the remote centralized Cisco CallManager by the WAN and PSTN.

Figure 8-1 Branch Office Cisco IP Phones Connected to the Remote Central Cisco CallManager

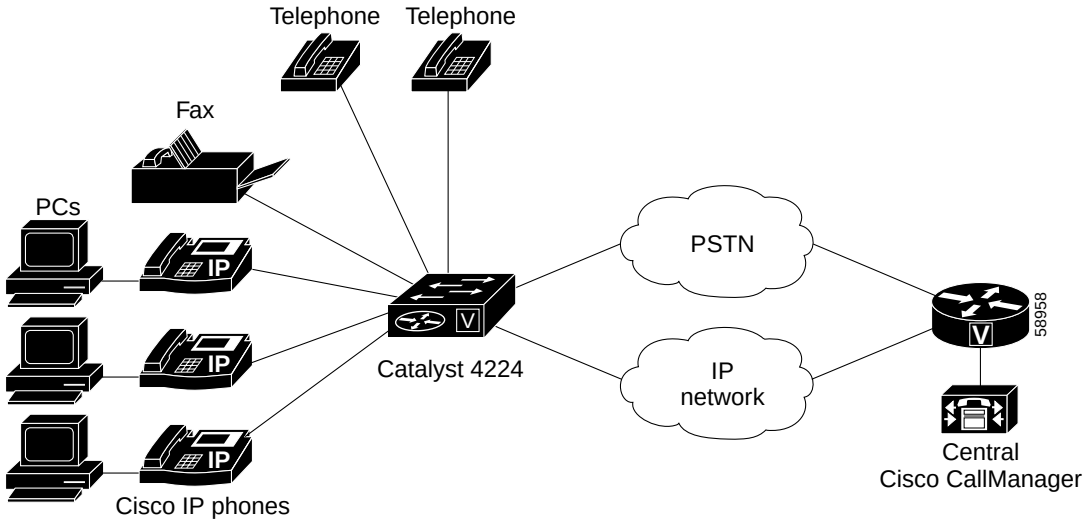
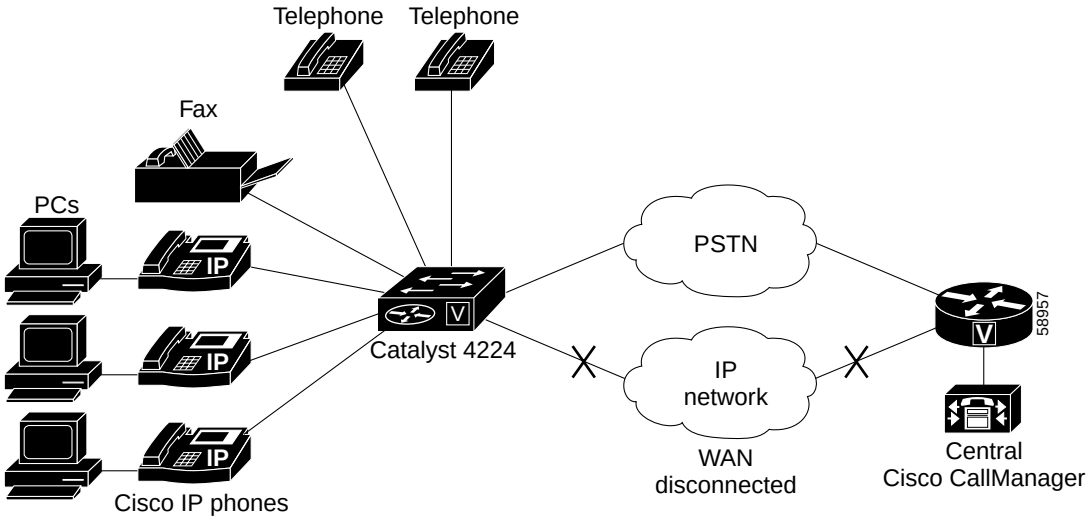


Figure 8-2 shows that the WAN connection to the branch office is down and that the Cisco IP Phones are able to make calls by being connected to the local SRST Catalyst 4224. This Catalyst 4224 acts as a fallback Cisco CallManager. The branch office Cisco IP Phones on the LAN network, connected to the PSTN, are capable of making off-net calls outside the network using the Catalyst 4224s.

Figure 8-2 Branch Office Cisco IP Phones Operating in Survivable Remote Site Telephony (SRST) Mode



Configuring Survivable Remote Site Telephony

When SRST is enabled, Cisco IP Phones do not need to be reconfigured individually during Cisco CallManager fallback mode because Cisco IP Phones retain the same configuration used with the primary Cisco CallManager.

To configure SRST on the Catalyst 4224, use the following steps, beginning in global configuration mode:

	Task	Command
Step 1	Enable SRST support and enters Cisco CallManager fallback mode.	`C4224(config)# call-manager-fallback`
Step 2	Enable the Catalyst 4224 to receive messages from the Cisco IP Phones through the specified IP addresses and ports.	`C4224(config-cm-fallback)# ip source-address` *ip-address* `port` *port*

	Task	Command
Step 3	Configure the allowable number of Cisco IP Phones that can be supported by the Catalyst 4224. **Note** You cannot reduce the allowable number of Cisco IP Phones once the maximum allowable number is configured, without rebooting the Catalyst 4224.	`C4224(config-cm-fallback)# max-ephones` *max phone*
Step 4	Set the allowable number of directory numbers that can be supported by the Catalyst 4224. **Note** You cannot reduce the allowable number of the directory numbers once the maximum allowable number is configured, without rebooting the Catalyst 4224.	`C4224(config-cm-fallback)# max-dn` *max directory number*
Step 5	(Optional) Configure the time interval between keepalive messages sent by Cisco IP Phones to the Catalyst 4224. E This setting is effective during Cisco CallManager fallback mode when SRST is enabled.	`C4224(config-cm-fallback)# keepalive` *seconds*
Step 6	(Optional) Assign a default destination number for incoming telephone calls.	`C4224(config-cm-fallback)# default-destination` *telephone number*
Step 7	(Optional) Create a global prefix that can be used to expand the abbreviated extension numbers into a fully qualified E.164 numbers. The **extension-length** keyword enables the system to convert a full E.164 telephone number back to an extension number for the purposes of caller ID display, received, and missed-call lists.	`C4224(config-cm-fallback)# dialplan-pattern` *tag pattern* `extension-length` *number*
Step 8	(Optional) Allow transfer of telephone calls by Cisco IP Phones to other phone numbers (IP and non-IP phone numbers).	`C4224(config-cm-fallback)# transfer-pattern` *transfer-pattern*

	Task	Command
Step 9	(Optional) Configure trunk access codes for each type of line—Basic Rate Interface (BRI), E&M, Foreign Exchange Office (FXO), and Primary Rate Interface (PRI)—so that the Cisco IP Phones can access the trunk lines during Cisco CallManager fallback mode when the SRST feature is enabled.	`C4224(config-cm-fallback)# access-code {bri | e&m |fxo | pri}`
Step 10	(Optional) Configure the telephone number that is speed-dialed when the message button on a Cisco IP Phone is pressed.	`C4224(config-cm-fallback)# voicemail` *phone-number*
Step 11	Exit Cisco CallManager fallback configuration mode.	`C4224(config-cm-fallback)# exit`
Step 12	Exit global configuration mode.	`C4224(config)# exit`

Note In Cisco CallManager fallback mode, huntstop is set by default. (Huntstop disables all further dial-peer hunting for the dial peers associated with the Cisco IP Phone lines if a call fails using hunt groups.) Use the **no huntstop** command only if you want to disable huntstop.

Verifying Survivable Remote Site Telephony

To verify that the SRST feature is enabled, follow these steps:

Step 1 Enter the **show run** command to verify the configuration.

Step 2 Enter the **show call-manager-fallback all** command to verify that the SRST feature is enabled.

Step 3 Verify that the default Catalyst 4224 IP address on the Cisco IP Phone is the same as the IP address of the SRST Catalyst 4224. Use the settings display on a Cisco IP Phone to do so.

Step 4 Temporarily block the TCP port 2000 Skinny Client Protocol (SCCP) connection for one of the Cisco IP Phones. This forces the Cisco IP Phone to lose its connection to Cisco CallManager and register with the SRS Catalyst 4224. Perform the following:

1. Use the appropriate **access-list** command to temporarily disconnect a Cisco IP Phone from Cisco CallManager.

Note The Cisco IP Phone 7960 and 7940 display the message "CM Fallback Service Operating" at the bottom of the display screen. The Cisco IP Phone 7910 displays the message "CM Fallback Service" for five seconds every 30 seconds. This message is brief because the Cisco IP Phone 7910 has a two-line display area and the display panel is also used to display the telephone number.

2. Delete the **access-list** command to restore normal service for the phone by entering the **no** form of the appropriate **access-list** command.
3. Use the **debug ephone register** command to observe the registration process of the Cisco IP Phone on the SRST Catalyst 4224.
4. You can also enter the **show ephone** command to display the Cisco IP Phones that have registered to the SRST Catalyst 4224.

Troubleshooting Survivable Remote Site Telephony

Use the following commands to troubleshoot the SRST feature.

Command	Purpose
`debug ephone keepalive`	Sets keepalive debugging for the Cisco IP Phone.
`debug ephone register`	Sets registration debugging for the Cisco IP Phone.
`debug ephone state`	Sets state debugging for the Cisco IP Phone.
`debug ephone detail`	Sets detail debugging for the Cisco IP Phone.
`debug ephone error`	Sets error debugging for the Cisco IP Phone.
`debug ephone statistics`	Sets call statistics debugging for the Cisco IP Phone.

Command	Purpose
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages.

Monitoring and Maintaining Survivable Remote Site Telephony

Use the following commands to monitor and maintain the Catalyst 4224 with the SRST feature:

Command	Purpose
C4224# **show run**	Displays the configuration.
C4224# **show call-manager-fallback all**	Displays the detailed configuration of all the Cisco IP Phones, voice ports, and dial peers of the SRST Catalyst 4224.
C4224# **show call-manager-fallback dial-peer**	Displays the output of the dial peers of the SRST Catalyst 4224.
C4224# **show call-manager-fallback ephone-dn**	Displays the Cisco IP Phone destination number.
C4224# **show call-manager-fallback voice-port**	Displays output for the voice ports.
C4224# **show ephone**	Displays Cisco IP Phone output.
C4224# **show ephone-dn**	Displays the Cisco IP Phone destination number.
C4224# **show ephone summary**	Displays a summary of the Cisco IP Phone output.
C4224# **show voice port summary**	Displays a summary of all voice ports.
C4224# **show dial-peer voice summary**	Displays a summary of all voice dial peers.

SRST Configuration Example

This section provides a configuration example for the SRST feature:

```
!
version 12.1
no service single-slot-reload-enable
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname Garfield
!
logging rate-limit console 10 except errors
!
ip subnet-zero
!
no ip finger
no ip domain-lookup
!
ip dhcp pool PHONE1
   host 10.1.0.2 255.255.0.0
   client-identifier 0100.3094.c337.cb
   option 150 ip 172.198.0.2
   default-C4224 10.1.0.1
!
ip dhcp pool PHONE2
   host 10.1.0.3 255.255.0.0
   client-identifier 0100.3094.c3f9.6a
   default-C4224 10.1.0.1
   option 150 ip 172.198.0.2
!
interface FastEthernet0/0
 ip address 10.1.0.1 255.255.0.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 ip address 172.198.0.1 255.255.0.0
 duplex auto
 speed auto
!
interface Ethernet1/0
 no ip address
 shutdown
 half-duplex
!
interface Serial1/0
```

```
 no ip address
!
interface TokenRing1/0
 no ip address
 shutdown
!
ip kerberos source-interface any
ip classless
no ip http server
!
snmp-server packetsize 4096
snmp-server manager
call rsvp-sync
!
mgcp modem passthrough voip mode ca
no mgcp timer receive-rtcp
!
mgcp profile default
!
dial-peer cor custom
!
call-manager-fallback
 ip source-address 10.0.0.1 port 2000 strict-match
 max-ephones 24
 max-dn 24
 dialplan-pattern 1 408734.... extension-length 4
 transfer-pattern 510650....
 voicemail 11111
!
line con 0
 transport input none
line aux 0
line vty 0 4
 login
!
end
```